IN THE CLAIMS:

Please amend the claims as follows:

1. (Currently Amended) A method for filtering comprising adaptive filtering an input signal, interpolating a filtered signal to provide an interpolated filtered signal, interpolating the input signal for adapting the adaptive filtering, and adapting properties of an interpolation of the filtered signal in response to an error signal indicative of a difference between a desired signal and the interpolated filtered signal.

2. (Cancelled)

3. (Currently Amended) A method for filtering comprising:

adaptive filtering an input signal;

interpolating a filtered signal;

interpolating the input signal for adapting the adaptive filtering,

adapting properties of an interpolation of the filtered signal;

providing a reference signal, and combining an interpolated filtered signal and the reference signal for forming an error signal; and

~~The method according to claim 2 comprising~~ adapting the properties of the interpolation according to the error signal and the interpolated filtered signal.

4. (Currently Amended) The method according to claim 1~~2~~ comprising adapting the properties of the interpolation by changing at least one coefficient of the interpolation.

5. (Previously Presented) The method according to claim 4 comprising adapting the at least one coefficient of the interpolation by using a normalized least mean square algorithm, wherein the method further comprises using the error signal and the interpolated filtered signal as inputs for the algorithm.

6. (Currently Amended) The method according to claim ~~2~~1 comprising:

a) computing the filtered signal by an equation

$y(n) = W^t(n)X(n)$;

b) computing the interpolated filtered signal by an equation

$Y_I(n) = I^t(n)Y(n)$;

c) adapting interpolation coefficients of an interpolator by an equation

$$\mathbf{I}(n+1) = \mathbf{I}(n) + \frac{\mu_I}{\varepsilon + \mathbf{Y}^t(n)\mathbf{Y}(n)} e(n)\mathbf{Y}(n)$$

where μ_I is a step-size used to adapt the coefficients of the interpolator, $e(n)$ is an output error, $I(n) = [i(n)_1, i(n)_2, \ldots, i(n)_M]^t$ is an $M \times 1$ vector containing the interpolation coefficients of the interpolator, $Y(n) = [y(n), y(n-1), \ldots, y(n-M+1)]^t$ is a vector of past M samples from the filtered signal $y(n)$, and ε is a constant;

d) computing the output error $e(n)$ by an equation

$e(n) = d(n) + z(n) - y_I(n)$;

e) computing a filtered input vector $X_I(n)$ by an equation

$$\mathbf{X}_I(n) = \sum_{j=0}^{M-1} i_j \mathbf{X}(n-j); \text{ and}$$

f) updating filtering weights by an equation

$$\mathbf{W}(n+1) = \mathbf{F}\{\mathbf{W}(n) + \mu e(n)\mathbf{X}_I(n)\} + \mathbf{q},$$

where $W^t(n)$ is indicative of a filter, X(n) is indicative of the input signal, d(n) is indicative of the desired signal, z(n) is indicative of noise, F{} indicates as a function of, and q is a constant.

7. (Previously Presented) The method according to claim 1 comprising using finite impulse response filtering in said adaptive filtering.

8. (Currently Amended) An apparatus comprising:

an adaptive filter for filtering an input signal;

a first interpolator for interpolating a filtered signal to provide an interpolated filtered signal;

a combiner that provides an error signal indicative of a difference between a desired signal and the interpolated filtered signal;

a second interpolator for interpolating the input signal, wherein an interpolated input signal is arranged to be used to adapt the adaptive filter; and

a first adapting block for adapting the properties of the first interpolator in response to the error signal.

9. (Cancelled)

10. (Currently Amended) An apparatus comprising:

an adaptive filter for filtering an input signal;

a first interpolator for interpolating a filtered signal;

a second interpolator for interpolating the input signal, wherein an interpolated input signal is arranged to be used to adapt the adaptive filter; and

a first adapting block for adapting the properties of the first interpolator;

an input for receiving a reference signal, and a combiner for combining an interpolated filtered signal and the reference signal for forming an error signal, ~~The apparatus according to claim 9,~~ wherein the properties are arranged to be adapted according to the error signal and an interpolated filtered signal.

11. (Currently Amended) The apparatus according to claim 8~~9~~, wherein the first adapting block is adapted to change at least one coefficient of the first interpolator.

12. (Previously Presented) The apparatus according to claim 11, wherein the first adapting block is adapted to use a normalized least mean square algorithm to adapt the at least one coefficient of the first interpolator, wherein the error signal and the interpolated filtered signal are arranged to be used as inputs for the algorithm.

13. (Previously Presented) The apparatus according to claim 8, also comprising a second adapting block for adapting properties of the adaptive filter.

14. (Previously Presented) The apparatus according to claim 8, wherein said adaptive filter is a FIR filter.